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In RF power and spectral efficient digital communication systems, the transmitted signals are required to have both narrow-band spectrum and constant envelope. To meet these requirements, it is desirable to use spectrally efficient modulation techniques. FQPSK [1-2] 'and 16-state Superposed QAM (16-SQAM [1], [3], also called FQAM) have been demonstrated to achieve higher Dower and spectral efficiency, and also show good error probabilities in NLA and Additive White Gaussian Noise (AWG.N) channels- These signals contain "offset" property in quadrature (Q) channel l relative to in-phase (I) channel, which prevents the envelopes of the modulated signals from crossing zeros and allows to achieve higher power efficiency [2].

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Final Technical Report for

Bandwidth Efficient Aeronautical Telemetry Modem and Transceiver Research

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INTRODUCTION

In RF power and spectral efficient digital communication systems, the transmitted signals are highly required to have both narrow-band spectrum and constant envelope. To meet these requirements, it is desirable to use spectrally efficient modulation techniques. FQPSK [1-2] and 16-state Superposed QAM (16-SQAM [1], [3], also called FQAM) have been demonstrated to achieve higher power and spectral efficiency, and also show good error probabilities in NLA and Additive White Gaussian Noise (AWGN) channels. These signals contain "offset" property in quadrature (Q) channel relative to in-phase (I) channel, which prevents the envelopes of the modulated signals from
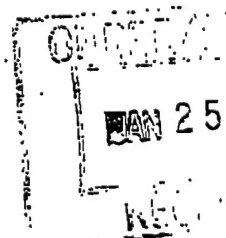
crossing zeros and allows to achieve higher power efficiency [2].

The spectral efficient modulations, however, are more vulnerable to frequency selective fading when they are transmitted through multipath propagation channels which introduce Intersymbol Interference (ISI) and degrade system performance. To improve system performance it is important for receivers to compensate for the distorted signals by using equalizer. Conventionally, the coefficients of adaptive equalizers are updated in startup period with the aid of a training sequence, which is known at both the transmitter and the receiver. However training sequence based equalization is not an attractive feature and may not be practical in many applications. So blind equalization can play a very important role in these applications.

Blind equalization relies solely on the equalized output signal and a *priori* statistical knowledge of the transmitted data constellation. Godard's algorithm (GA) [4], which belongs to CMA [5], for complex two-dimensional data communication systems is the most widely referenced technique in both industry and academia due to its simple and easy to be implemented in digital signal processing (DSP) chip. However, blind equalization algorithm converges very slowly compared to the training sequence based equalization which employs the *least-mean-square* (LMS) algorithm. In order to speed up the convergence process, a variety of improved techniques based on Godard's algorithm have been appeared in the literature, but most of them are applied to traditional QAM, or non-offset QAM.

In this paper, we consider a class of *offset* QAM modulations as our target to study the application of blind channel equalization. Specifically, we present an efficient method to create error signal by alternatively sampling the equalized output signal at a bit rate of T_b between I and Q channels. Then we use the well-known *constant modules algorithm* (CMA) [4] to achieve a fast convergence speed for *offset* QAM modems. The goal of this paper is to investigate how the blind equalization using CMA method can be applied to the NLA RF power efficient FQPSK and FQAM systems.

FQPSK AND FQAM SYSTEMS

A. FQPSK Signals
FQPSK systems have been described in numerous references [1-2], [6], [8]. In the family of these systems, FQPSK-B, a constant envelope modulation with baseband waveform-shaped and cross-correlated between I and Q channels, has been successfully demonstrated to have both power and spectral efficiency and good bit error rate (BER) performance for several applications.

B. FQAM Signals
One of FQAM's is a 16-state offset QAM modulation and could be constructed by parallel-type with two independent FQAM (previously designed as SQAM) modulators [1], [3]. A block diagram of FQAM modulator is shown in Fig. 1. Input NRZ data $D(t)$ enter a serial-to-parallel converter (S/P) to be split into four parallel NRZ sequences $I_1(t)$, $Q_1(t)$, $I_2(t)$, and $Q_2(t)$, which go to two FQAM modulators labeled by #1 and #2, respectively. These two FQAM modulators are identical to each other. The modulated signals $Y_1(t)$ and $Y_2(t)$ pass through hardlimiters, which approximate the characteristics of a fully saturated amplifier and also remove envelope fluctuation

of the modulated signal such that the outputs of $Z_1(t)$ and $Z_2(t)$ do not suffer further degradation from non-linear amplifications. The output voltage level of HPA1 is twice that of HPA2, that is, the saturated output power of HPA1 is 6 dB higher than that of HPA2. Finally, a modulated FQAM signal is represented as

$$s(t) = \sum_k \{a_k p(t - kT_s)\cos(2\pi f_c t) + b_k p(t - (k - 1/2)T_s)\sin(2\pi f_c t)\} \qquad (1)$$

where a_k, $b_k = \pm 1, \pm 3$, independent and equiprobable. T_s is symbol duration and $T_s = 4T_b$. $p(t)$ is pulse signal of FQAM defined as [3].

Fig. 2 shows power spectrum density (PSD) of FQAM signal passing through a hardlimited non-linear channel. Conventional 16-QAM signals cannot operate in a non-linear channel without significant spectral distortion while FQAM signals may operate in it without significant spectral distortion. Symbol Error Rate (SER) of FQAM through hardlimited and AWGN channels is given in Fig. 3, where its performance degradation is about 0.8 dB at $P(e)=10^{-4}$ compared to ideal 16-QAM signal in a linear channel.

BLIND EQUALIZATION FOR OFFSET MODEMS

A mathematical structure of adaptive equalizer for offset QAM and PSK is depicted in Fig. 4. The sampled complex input signal $v(nT_s/2)$ is fed to a feedforward (FF) finite impulse response (FIR) filter with N taps spaced $T_s/2$ delay line in a part of fractionally-spaced equalizer. The delay signals are weighted by $\{W_i, i=0,1,....,N-1\}$ and summed. Then the summed signal is subtracted by the other summed signal from the feedback (FB) FIR filter part to produce the equalized signal $\hat{I}_k$. The equalized signal is split into the real and imaginary parts corresponding to the signals in I and Q channels. The quantization provides the basic decision detection. In the case of $M=4$, such as OQPSK, FQPSK and GMSK, a simple binary decision device is modeled. In the case of M=16, such as 16-QAM and FQAM, a 4-level quantizer is used. A equalizer is called as linear equalizer (LE) if there is no feedback FIR part, otherwise called as decision feedback equalizer (DFE).

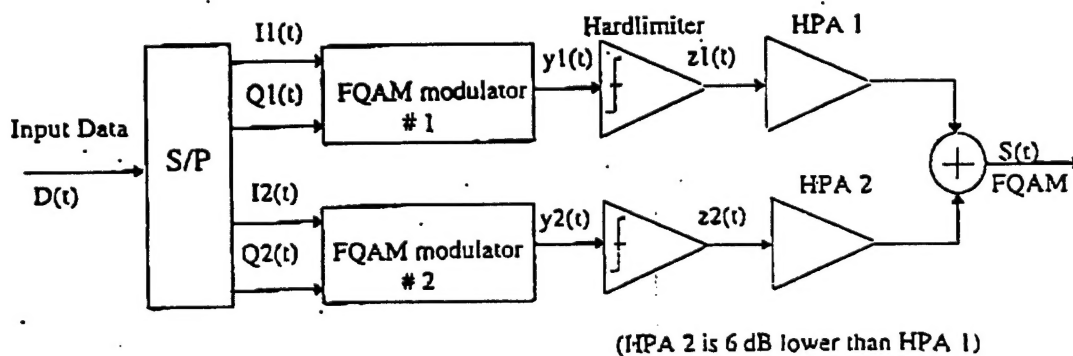(HPA 2 is 6 dB lower than HPA 1)

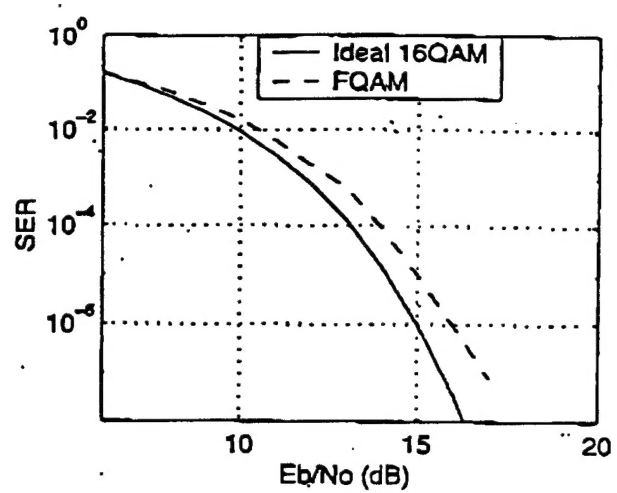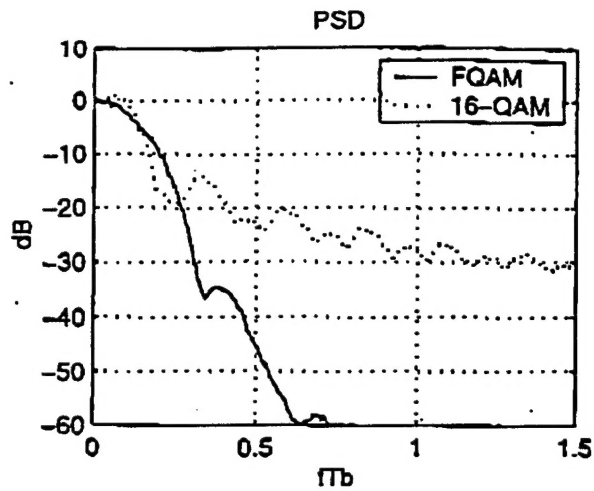Fig. 1. Block diagram of parallel-type FQAM modulator [1], [3].

Fig. 2. PSD of FQAM in hardlimited NLA channel. Fig. 3. SER of FQAM in hardlimited NLA channel.
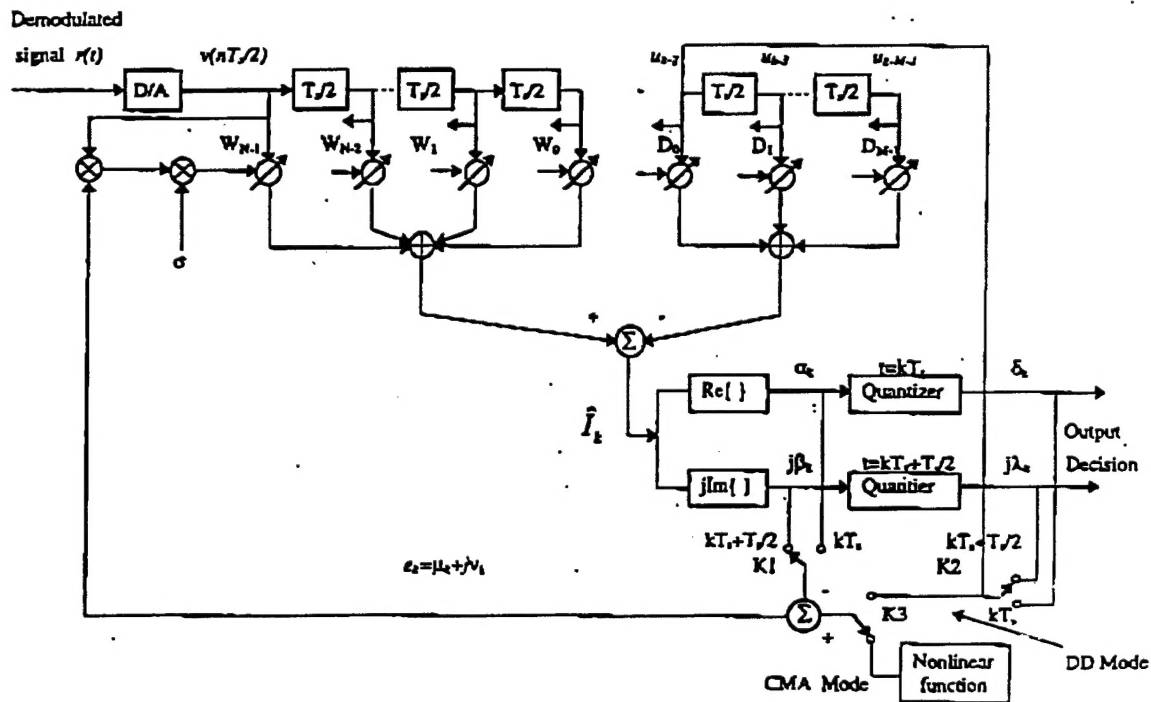Ideal 16-QAM in linear channel.

Fig. 4. A mathematical structure of decision feedback complex equalizer for offset QAM
where $\hat{I}_k = \{\alpha_k + j\beta_k\}$ is the output signal of the equalizer.

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A. Blind Equalization for FQPSK

In the adaptive equalization, the coefficients of the equalizer are carried out by minimizing *mean square error* (MSE) which is difference between the equalizer output and the desired data symbol. Unfortunately, in the case of blind equalization, the desired symbol is unknown. CMA blind approach is to minimize a cost function whose minimum is equivalent to minimizing MSE and depends on the output of the equalizer and a priori knowledge of statistics of the transmitted data constellation. A general cost function proposed by Godard [4] is

$$D^p = E(\left|\hat{I}_k\right|^p - R_p)^2 \tag{2}$$

where p is a positive and real integer, R_p is a positive real constant. Minimization of D^p with respect to the equalizer coefficients can result in a coefficient update equation. Of particular importance is the case $p=2$, which leads to the relatively simple algorithm called as Godard's algorithm (GA) and fast convergence speed compared with $p=1$

$$C_{k+1} = C_k + \mu V_k^* \hat{I}_k (R_2 - \left|\hat{I}_k\right|^2) \tag{3}$$

with

$$R_2 = \frac{E(\left|I_k\right|^4)}{E(\left|I_k\right|^2)} \tag{4}$$

where $I_k = a_k + jb_k$ is kth symbol in the transmitter, μ is the step size parameter and the superscript * denotes complex conjugate. In the trained equalization, the cost function is

$$CF = E(\left|I_k - \hat{I}_k\right|^2) \tag{5}$$

Minimization of CF with respect to the equalizer coefficients leads to LMS algorithm

$$C_{k+1} = C_k + \mu V_k^* (I_k - \hat{I}_k) \tag{6}$$

where μ is the step-size parameter for LMS, I_k is the desired symbol sequence which may be either the trained sequence in startup period or the decision-directed (DD) sequence in tracking period. In startup period, I_k is known by the receiver and is replaced by decision-directed value of $\mathrm{sgn}(\hat{I}_k)$ in tracking period because the eye is open. Compared (3) with (6), the update equation can be written in the following general form

$$C_{k+1} = C_k + \mu e_k V_k^* \tag{7}$$

where the expression of the error signal e_k for the CMA and the LMS follows immediately from (3) and (6), respectively.

For the envelope normalized symbol constellations, or maximum value in I or Q channel is normalized to 1, R_2 is equal to 1 for FQPSK due to its constant envelopes. In the modified CMA, the error signal e_k is created by alternatively sampling the equalized signal $\hat{I}_k$ between I and Q channels and is expressed as

$$e_k = \hat{I}_k (R_2 - |\hat{I}_k|^2)$$

$$= \begin{cases} \alpha_k (1 - \alpha_k^2), & t = kT_s \\ j\beta_k (1 - \beta_k^2), & t = kT_s + T_s/2 \end{cases} \tag{8}$$

Unlike QPSK, where the error signal e_k always takes a complex value, the error signal e_k for offset QPSK may be either real or imaginary, depending on a certain time t. In the case of trained equalization, where LMS algorithm is used, the error signal e_k is

$$e_k = (I_k - \hat{I}_k)$$

$$= \begin{cases} a_k - \alpha_k, & t = kT_s \\ j(b_k - \beta_k), & t = kT_s + T_s/2 \end{cases} \tag{9}$$

In general, the error signal in the blind equalization goes to zero very slowly than one in the trained equalization so that blind algorithm converges slowly than LMS algorithm.

B. Blind Equalization for FQAM

FQAM could be modeled as an offset 16-QAM with Feher's patented processors [1]. Being considered constellation at sampling points, FQAM data constellation is depicted in Fig. 5. In CMA, the constant R is a function of the statistics of the symbols in the data constellation.   CMA
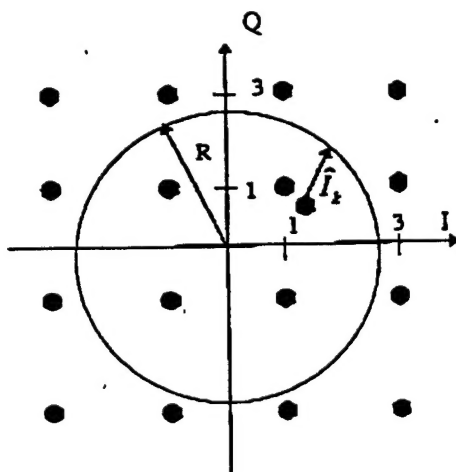

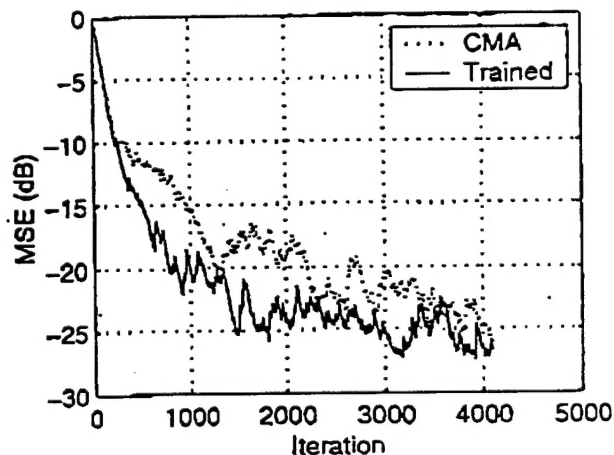Fig. 5. Interpretation of constant R as radius in FQAM.

Fig. 6. Learning curves for NLA RF power efficient FQPSK. LE: FF=21 taps, E_b/N_o=20 dB.

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minimizes the dispersion of the equalizer's output samples $\hat{I}_k$ around a circle of R.

It should be noted that, even in the absence of noise, the error signal e_k generally does not go to zero. This is because all tracks of points on constellation can not be represented by a constant circle of R. For the constellation of FQAM shown in Fig. 5, R_2 equals to 8.2 by using equation (4). Compared to trained equalizer, which employs the LMS algorithm to update equalizer's coefficients, blind equalization algorithm converges very slowly. Like a blind equalizer for offset FQPSK, a blind equalizer for FQAM should be switched to the decision-directed (DD) equalization mode once the eye signals are open in order to speed up the convergence process and reduce the error signal e_k in tracking period.

COMPUTER SIMULATIONS

A. Multipath fading model

There is one frequently referenced multipath fading model in line-of-sight microwave and aeronautical communications: *three-path model* [7]. Three-path fading model describes the multipath propagation in terms of a primary ray and a dominant interference ray, where a primary ray consists of two rays with a 180 phase degree between them and is characterized as a flat attenuation. Rummler used the simplest form of mathematical modeling function whose impulse response is

$$h(k) = a\delta(k) - abe^{j\theta}\delta(k-\tau) \tag{10}$$

where the parameters a and b control a flat loss term and the relative depth of the fading, respectively. Phase shift $\theta = \omega_n \tau$ and is chosen to set a null position from the signal carrier. ω_n, called as the notch frequency, is the angular frequency of fade minimum measured from the band center. τ is the delay difference between primary and dominant rays and usually fixed at 6.3 ns, which has been accepted by many researchers but may be chosen as any convenient value.

B. Blind Equalization for FQPSK

In our simulation, FQPSK modulated IF signals are assumed to be transmitted over multipath channels corrupted by AWGN. An IF model is chosen for a convenience to deal with an asymmetrical notch location from the signal center frequency. In the receiver, the received signal faded by multipath propagation is first coherently demodulated by a phase-locked carrier signal, then passes through Butterworth low-pass filter. The input signal to the equalizer is sampled at bit rate of $T/2$, and fed to a fractionally-spaced blind equalizer. Blind equalizer is initialized to be all zeros except for the unity center coefficient. In the simulation, a LE is used in start-up period and DFE is switched in tracking period.

First we compare CMA with LMS algorithm for NLA FQPSK. Fig. 6 shows the simulation results of MSE. As we can see that the convergence rate of CMA is not much slower than that of trained LMS. The received eye diagrams are given in Fig. 7. In Fig. 6 and Fig. 7, the parameters
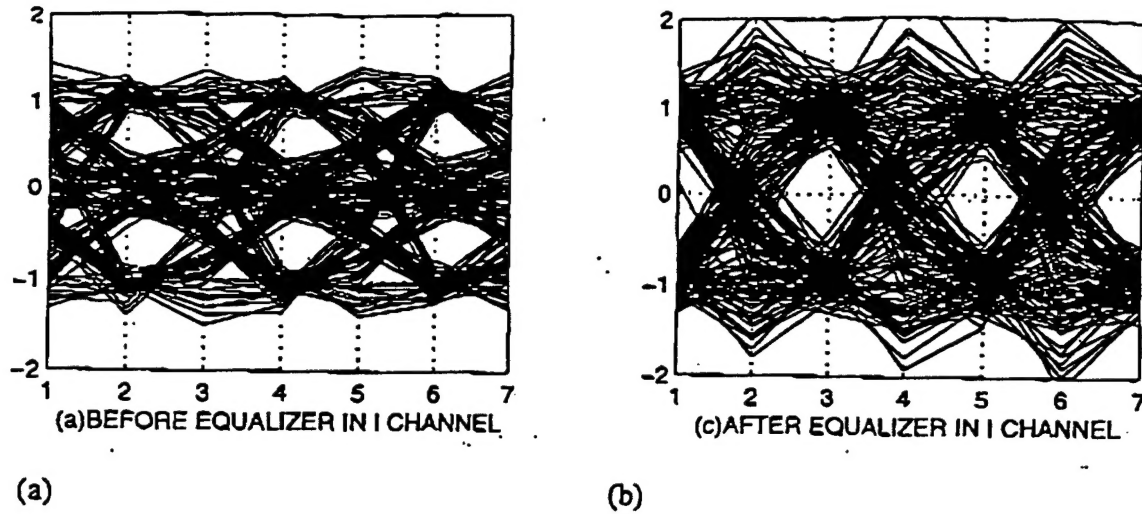
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(a)BEFORE EQUALIZER IN I CHANNEL

(a)

(c)AFTER EQUALIZER IN I CHANNEL

(b)

Fig. 7. Received eye diagrams of RF faded FQPSK signals. Notch is set at edge of bandwidth and its depth is 16 dB, τ= 9 samples. LE: FF=21 taps. E_b/N_o=20 dB. (a). Before equalizer. (b) After blind equalizer.
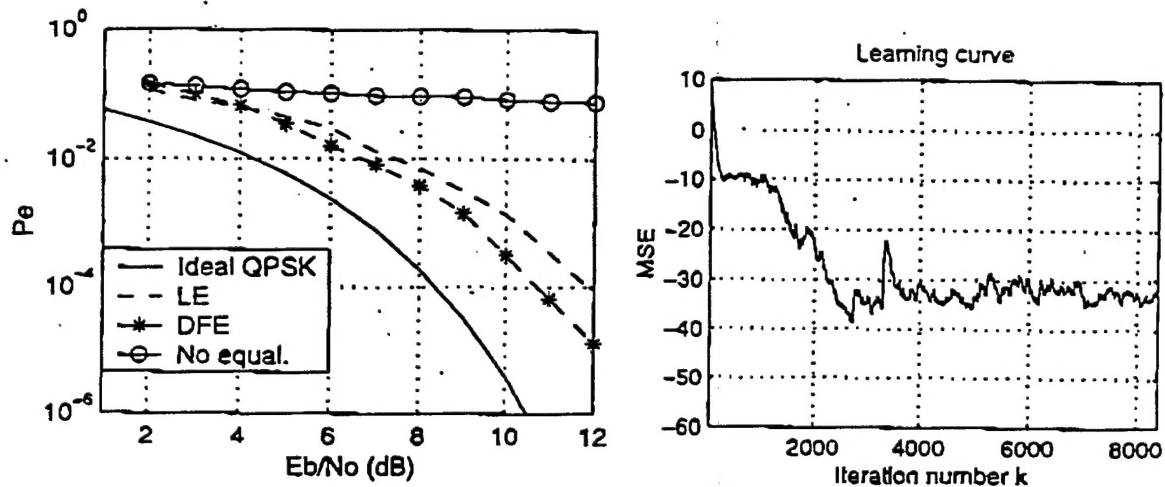

Fig. 8. BER performance of NLA FQPSK. Notch depth is 16 dB, LE: FF=10, DFE: FF=5, FB=5.

Fig. 9. Learning curve of adaptive equalizer for NLA-FQAM in dynamic fading channel.

of both a and b are set to 0.8, corresponding to a notch depth of 16 dB. τ is set to 9 samples away from the primary ray (one sample per bit), and phase shift θ is chosen to set a null at the edge of bandwidth, which corresponds to the worst case due to an asymmetrical distortion (see Fig. 11). BER of NLA FQPSK is simulated in DD mode after eye signals are open and shown in Fig. 8. where DFE outperforms LE due to avoiding noise enhancement problems.

C. Blind Equalization for FQAM

Fig. 9 shows the simulation result of learning curve for NLA FQAM system in a dynamic fading channel where changing speed of the notch position is set to $f_s/128$ by using CMA and LMS algorithm, where f_s is symbol rate. Like blind equalization for FQPSK system, the operation mode should be switched to DD mode from CMA mode once eye signals are open, which corresponds to that MSE is less than -15 dB. Also after DD mode is chosen, a feedback part of equalizer is added to adaptive equalizer in order to cancel "postcursor ISI" caused by past data symbols. It can be seen from Fig. 9 that the coefficients of DFE are adaptively updated in DD mode to track channel changing after CMA mode is switched to DD mode at iterations (symbols) of 2048. Constellations of the received FQAM signals through NLA and dynamic fading channels at the equalizer's input and output are given in Fig. 10, where change rate of notch position is set to $f_s/128$.

To evaluate the performance degradation vs. the notch position, SER curves of FQAM using LE only at $E_b/N_o = 15$ dB are obtained by setting notch depth=12 dB (a=0.8; b=0.6), and varying ω_n. It can be seen from Fig. 11 that the notch position has much more effects on the SER around the edge-band ($f_s T_s$=0.6). Note that 3 dB bandwidth of the receiver is normalized to $B_i T_s$=0.53, where B_i is a 3 dB bandwidth and T_s is symbol duration.

CONCLUSIONS

An efficient blind equalization of NLA power efficient systems for FQPSK and FQAM is presented in multipath frequency selective fading environment. By employing offset sampling strategy at the equalizer's output, a modified CMA with fast convergence speed has been developed for NLA offset QAM. Computer simulation using the CMA demonstrated the great improvement in opening eye signals for FQPSK and FQAM systems.
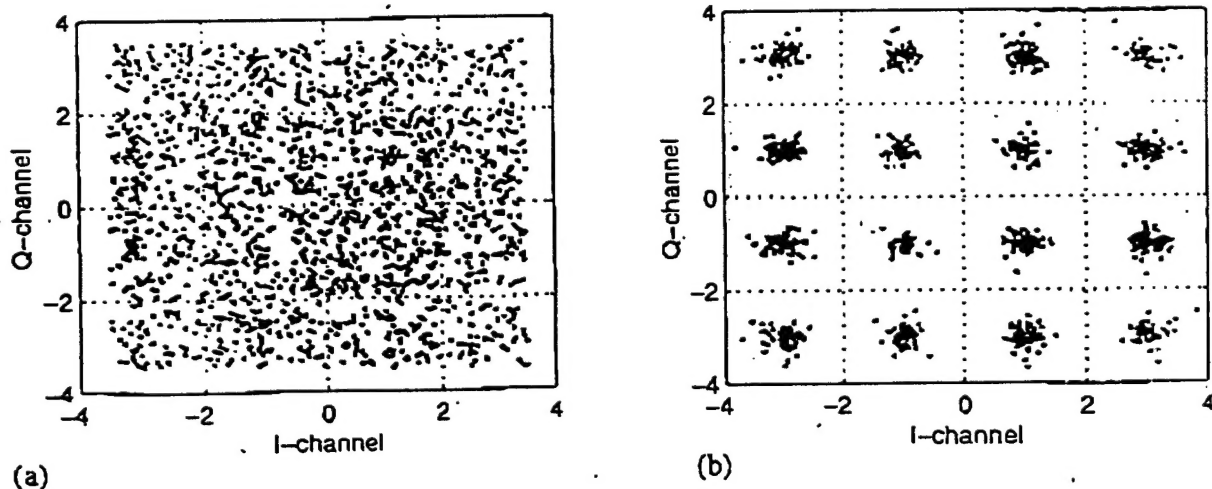

(a)   (b)

Fig. 10. Constellations of FQAM in the receiver through a dynamic fading channel. Notch depth is 16 dB, FF=15 taps, FB=15 taps. E_b/N_o=25 dB. (a). Before equalizer. (b) After equalizer.

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FQPSK Consortium-Digcom, Inc., c/o Dr. Feher; 44685 Country Club Dr.; El Macero, CA 95618; Tel. 530-753-0738; Fax 530-753-1788; E-mail feherk@yahoo.com

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